heard by the user on the side of the voice communications device 61 through the network 15.

The dual-talk detector 64 in the voice communications device 62 simultaneously monitors the voice signal EV that is the input of the voice encoder and the voice signal DV that is the output of the voice decoder 17, thereafter totals the periods (dual-talk time) in which two paths (i.e., two voice signals EV and DV) are simultaneously in a state of voice presence, thereafter detects whether the dual-talk time has a tendency to become long or become short, and outputs a control signal C11 that corresponds to the detection result.

Specifically, the dual-talk time for the past ten seconds, for example, is measured at intervals of one second, and, if the dual-talk time is longer than that at the pre-measuring time (i.e., one second ago), the control signal C11 by which the higher threshold TH1 becomes small by one packet is output.

In contrast, if the dual-talk time is shorter than that at the pre-measuring time, the control signal C11 by which the higher threshold TH1 to be output becomes large by one packet is output.

The queue length detector 30A in the buffer device 66 shown in Fig. 10 receives this control signal C11. The queue length detector 30A first changes the higher threshold TH1 in accordance with the control signal C11, the detector 30A then examines whether the queue length is

longer or shorter than the changed higher threshold TH1, and, if longer, the detector 30A switches the control signal C3 from the inactive state to the active state.

For example, in Fig. 10, the higher threshold TH1 is set at the position that corresponds to the queue length in the middle of 100 packets and 99 packets. If this is changed to become larger by one packet, the higher threshold TH1 is moved to the position that corresponds to the queue length in the middle of 101 packets and 100 packets. Subsequently, the relationship between the queue length and the higher threshold TH1 is examined on the basis of a changed higher threshold TH, and the packet deletion is executed for the first time when the 101st voice packet P101 is stored.

(C-2) Effect of the third embodiment

According to this embodiment, the same effect as in the first embodiment can be obtained.

In addition, in this embodiment, since the fixed delay (i.e., higher threshold TH1) can be dynamically changed in accordance with conversation patterns, a delay can be reduced when a dual-talk state that easily gives an unpleasant feeling to users frequently occurs, and thereby the quality of communication can be improved.

Contrarily, in a state (non-dual-talk state) where, for example, voice guidance is flowing, the fixed delay is enlarged, and this enlargement can make it difficult to generate a voice interruption caused by a buffer shortage.

(D) Fourth Embodiment

Only the difference between this embodiment and the first and third embodiments will be described hereinafter.

(D-1) Structure and operation of the fourth embodiment

A structure of a principal part of the voice communications system 70 of this embodiment is shown in Fig. 11.

In Fig. 11, the functions of each component and each signal, to which the same reference characters as that of Fig. 9 are given, are the same as those of Fig. 9.

That is, the voice communications system 70 of this embodiment has a structure in which the dual-talk detector 64 in the voice communications device 62 of the voice communications system 60 of the third embodiment is substituted with a power-variation-difference computing unit 71 and a zero-crossing counter 72.

With this structure, in the third embodiment, the control signal C11 is changed in accordance with the analytic result of a conversation pattern that corresponds to the tendency of extension/contraction of the dual-talk duration. On the other hand, in this embodiment, the control signal C11 is changed in accordance with the analytic result of a conversation pattern that corresponds to the time interval of an alternation in a conversation.

The computing unit 71 of this embodiment calculates voice power during a fixed time that has passed for the voice signal EV on the side of the encoder and the voice